Method For ~~Combating Ingress And Impulse Noise Using~~ Coded Modulation

This application claims priority under 35 USC §119(e)(1) of Provisional Application Number 60/117,962 filed January 29, 1999.

TECHNICAL FIELD OF THE INVENTION

The present invention relates to a method for combating ingress and impulse noise using coded modulation, and more particularly, to a system of communications for a CATV upstream plant using TDMA QAM modulation and a Bit-Interleaved Coded Modulation (BICM).

BACKGROUND OF THE INVENTION

Impulse and burst noise is a major impairment of the upstream channel of many CATV plants. A few technologies have been proposed to combat this noise, including Reed-Solomon coding, S-CDMA modulation, and OFDM (DMT) modulation.

BICM has been proposed by Zehavi and by Bigliery, as a technology for combating signal fading, which is a major impairment of wireless channels.

SUMMARY OF THE INVENTION

The present invention provides a coding method and a class of decoding methods that allow high throughput and high robustness digital communications over channels that are contaminated by impulse noise, as well as white and colored additive noise (noise that is characterized by heavy-tailed distribution), phase noise and signal fading. The class of decoding methods includes a low-complexity, low- delay method, and an iterative method, which is capable of lower error rate but with a higher complexity and higher delay.

This method is particularly useful in communications over CATV (cable TV) channels in the upstream direction, and a manner of using the method within a modulator previously proposed in the DOCSIS specification is provided.

The bit-interleaved coded modulation (BICM) encoding method of the present invention uses a conventional convolution encoder, a bit interleaving matrix and a conventional QAM mapper. The BICM encoder can be concatenated with an outer Reed-Solomon encoder.

The method of the present invention is highly robust to impulsive noise since the all the data bits are protected by the encoder, and due to the time diversity that is inserted into the signal as a result of the interleaving operation. The performance of this method in the presence of AWGN when using the methods described herein, combined with a Reed-Solomon encoder/decoder is better by 1-5 dB than the performance when using only a Reed-Solomon encoder/decoder.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 depicts a block diagram of the transmitter of the present invention.

Figure 2 depicts a block diagram of the non-iterative BICM decoder of a receiver of the present invention.

Figure 3 illustrates a score calculation technique for the case of 16QAM:

Figure 4 depicts the blocks effected by adding BICM to an FEC.

Figure 5 is a DOCSIS 1-2 upstream transmission processing scheme modified in accordance with the teachings of the present invention.

Figure 6 shows a BICM bit encoder having a rate ½ convolution encoder followed by a puncturing unit.

Figure 7 is a bit interleaving example.

DETAILED DESCRIPTION

The present invention provides a coding method and a class of decoding methods that provide methods for high throughput and high robustness digital communications over channels that are contaminated by impulse noise, as well as white and colored additive noise (noise that is characterized by heavy-tailed distribution), phase noise and signal fading. The class of decoding methods includes a low-complexity, low- delay method, and an iterative method, which is capable of lower error rate but with a higher complexity and higher delay.

This method is particularly useful in communications over CATV (cable TV) channels in the upstream direction, and a manner of using the method within a modulator previously proposed in the DOCSIS specification is provided.

The bit-interleaved coded modulation (BICM) encoding method of the present invention uses a conventional convolution encoder, a bit interleaving matrix and a conventional QAM mapper, as depicted in Figure 1. The BICM encoder can also be concatenated with an outer Reed-Solomon encoder.

The method of the present invention is highly robust to impulsive noise since the all the data bits are protected by the encoder, and due to the time diversity that is inserted into the signal as a result of the interleaving operation. The performance of this method in the presence of AWGN when using the methods described herein, combined with a Reed-Solomon encoder is better by 1-5 dB than the performance when using only a Reed-Solomon encoder.

In a presently preferred embodiment, the convolution encoder is a preferably punctured rate ½ convolution encoder with k=7 constraint length and generating polynomial 171 octal and 133 octal, as depicted in Figure 6. The coded bits are used to fill the rows of the interleaver matrix and are read column wise, as depicted in Figure 7. The preferred mapper uses Gray code mapping. Before the first data bit, the convolution encoder is initialized to a pre-determined state (e.g. all zeros). After the last data bit, the state of the encoder is brought to a pre-determined value by feeding dummy data bits into the encoder (e.g. feeding six zero bits).

Alternatives to the preferred embodiment are:

1. The encoder may employ multiple convolution encoders, wherein each encoder generates a portion of the input bits to the interleaver memory. For example, the odd lines and the even lines of the interleaver can be decoded by two different encoders. Another example - each bit of the QAM symbols can be generated by a different encoder. This allows for the use of different convolution encoders.
2. An alternative interleaving algorithm is random-address interleaving (i.e. the data bits are written serially and read in a non-regular manner, e.g. the address is generated by a random number generator).

Receiver methods: The BICM decoder of the present invention may be implemented in several ways. The present invention describes two decoding algorithms: A simple non-iterative method depicted in Figure 2,and an iterative method.

The Non-Iterative method is a computationally efficient approximation of the MLSE algorithm for the case of AWGN. The complexity of this decoding method is equivalent to the complexity of convolutional decoder modules that are used in various low cost IC's, such as DBS receiver's IC's.

The performance of the method has been compared to the performance of the MLSE method for a few simple scenarios, such as the case of AWGN channel and non-interleaved signal, and the difference in performance was less than 1 dB in the noise level. In the case of QPSK modulation, the method is an MLSE method.

The method is very similar to the conventional Viterbi method for convolutionally encoded QPSK or BPSK signals. As depicted in Figure 2, it is implemented by three units:

A Score Calculation Unit outputs the approximated scores of each channel bit.

Due to Gray code mapping, the real part and the imaginary part of the QAM symbols can be decoupled into two independent real valued PAM symbols, where each channel bit effects only one such PAM symbol. Thus, the approximated score of the channel bit b given "0" value, denoted by $L_0(b)$, is the square of the distance of the real or imaginary component of the decoder input to the nearest PAM symbol having a "0" value in the corresponding bit position. Similarly,

$L_1(b)$ is the squared distance to the nearest PAM symbol having a "1" value in the corresponding bit position. Figure 3 depicts a scoring example for a 16 QAM constellation map.

The scores are than clipped by a given threshold (i.e. if the score is higher than the threshold then the value of the threshold is assigned to the score). The threshold value may be an adaptive parameter. It set to be *m* times the variance of the noise, and *m* is increased to a high value (e.g. 25) when the noise is identified as a Gaussian process, and decreased to a low value (e.g. 4) when the noise is identified as a non-Gaussian process with heavy tailed distribution. Thus, the receiver may employ devices for tracking the noise variance and the tail of its distribution (e.g. measuring Kurtosis or the percentage of time the noise power is above 9 and 25 times its variance)

There are alternative computationally efficient approaches to approximate the score. One alternative approach is to use a one-dimensional look-up table whose input is the real or imaginary part of the decoder input.

Denoting the labels of the QAM symbols by ABCD, and assuming the score threshold is high, so that score-clipping does not occur, the approximated scores of the channel bits A, B, C, and D are:

$$L_1(A) = d^2, \quad L_0(A) = c^2,$$
$$L_1(B) = f^2, \quad L_0(B) = e^2,$$
$$L_1(C) = a^2, \quad L_0(C) = b^2,$$
$$L_1(D) = g^2, \quad L_0(D) = h^2.$$

As depicted in Figure 2, the scoring for each bit is deinterleaved using the Bit Deinterleaver Matrix. The matrix is filled column-wise and read row-wise.

Next, cnvolution decoding (Viterbi Algorithm Unit) is performed; the unit implements a method which is *exactly identical* to the well known 64 states Viterbi algorithm for soft decoding of convolutionally encoded BPSK (or QPSK) signal, except for a single difference: It uses the approximated scores calculated by the score calculation unit, rather than the conventional scores (which are the squared distances from the "0" and "1" levels of the binary signal). These methods implement 128 branches per bit for rate ½ code, and possibly less than that for

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punctured codes. There are several low cost IC's that implement such methods for data rates of more than 30 million information bits per second (e.g. DBS receivers). The cost of these IC's also includes A/D conversion, re-sampling, filtering, acquisition, de-interleaving, and Reed-Solomon decoding is below $10, and their power consumption is in the range of 1W or below. (See e.g., "*Digital Communications*", J.G. Proakis, 3rd Edition, McGraw Hill, 1995, pp. 483-486 for a description of the conventional algorithm).

The iterative method allows for significant improvement of the error rate of the decoder by:

1. Iterations between the most significant bits and the least significant bits of the channel symbols (soft outputs).

For this approach, soft decisions on bits of a certain channel symbol that have been decoded in this iteration or in previous iterations, are fed into the score calculation unit for decoding of other bits of this channel symbol. The soft decision can be obtained by multiple methods known in the literature, e.g. the SOVA algorithm (see, e.g. Hagenhauer).

2. Identification of noise impulse periods and giving low weights to the scores of the bits which occurred during this period.

Noise burst periods can be identified by:

a. Appearance of high magnitudes of the IF signal during a certain period of time.
b. Appearance of decoder inputs that are remote from all the constellation symbols during a certain period of time.
c. Low fidelity of the decoder decision (e.g. high residual decoder errors, or events where the state of the decoder having the best cost is not a parent of the state of the decoder having the best cost in the next iteration) relating to bit transmitted in a certain period of time.
d. Error correction of the outer Reed-Solomon decoder in data bytes that related to a certain period of time.

The decoder may employ a device that identifies relatively high rate of appearance of the above events (or other indication for impulse noise) and uses that to assess the likelihood that a certain time period is infected by high level of noise. This assessment is then provided to the decoder for giving low weights to the scores of the bits, which occurred during this period. Note

that indications a and b can be measured before the first iteration of the method, and thus can also be used in a non-iterative method.

The present invention provides a BICM method for combating impulse noise and provides improved communications methods.

Thus, the present invention provides:

1. Modification to an encoding method to make it suitable for use in an impulse noise environment.
2. Decoding methods that are computationally efficient and are robust to impulse noise.
3. Systems for communications over the CATV upstream plant using TDMA QAM modulation and BICM code.

An Appendix is attached that provides details for implementing a presently preferred embodiment of the present invention. This embodiment is for use in a DOCSIS 1.2 system and describes the changes to the DOCSIS specification needed to implement this embodiment.